CLAIMS

What is claimed is:

1. A method for suppressing noise from a speech signal, said method comprising:

obtaining an input speech signal;

performing linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;

computing spectrum tilt and noise-to-signal ratio (NSR) of said z-domain representation of said input speech signal;

obtaining spectrum tilt of a background noise model;

applying a gain to reduce energy of said input speech signal when said NSR is high;

reducing the spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is equivalent to said spectrum tilt of said background noise; and

applying an inverse filter to said input speech signal when said spectrum tilt of said input speech signal is not equivalent to said spectrum tilt of said background noise, wherein said inverse filter is an inverse of said z-domain representation of said background noise.

2. The method of claim 1, wherein said input speech signal comprises a plurality of sub-frames processed in sequence.

3. The method of claim 1, wherein said gain is adaptively based on characteristics of said input speech.

4. The method of claim 1, wherein said background noise model is a first order model.

5. A computer program product comprising:

a computer usable medium having computer readable program code embodied therein for suppressing noise from a speech signal; said computer readable program code configured to cause a computer to:

obtain an input speech signal;

perform linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;

compute spectrum tilt and noise-to-signal ratio (NSR) of said z-domain representation of said input signal;

obtain spectrum tilt of a background noise model;

apply a gain to reduce energy of said input speech signal when said NSR is high;

reducing the spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is equivalent to said spectrum tilt of said background noise; and

apply an inverse filter to said input speech signal when said spectrum tilt of said input speech signal is not equivalent to said spectrum tilt of said background noise, wherein said inverse filter is an inverse of said z-domain representation of said background noise.

6. The computer program product of claim 5, wherein said input speech

signal comprises a plurality of sub-frames processed in sequence.

7. The computer program product of claim 5, wherein said gain is adaptively based on characteristics of said input speech.

8. The computer program product of claim 5, wherein said background noise model is a first order model.

9. An apparatus for suppressing noise from a speech signal, said apparatus comprising:

an object for receiving an input speech signal;

an object for performing linear predictive coding (LPC) analysis on said input speech signal to obtain a z-domain representation of said input speech signal;

an object for computing spectrum tilt and noise-to-signal ratio (NSR) of said z-domain representation of said input signal;

an object for obtaining spectrum tilt of a background noise model;

an object for applying a gain to reduce energy of said input speech signal when said NSR is high ;

reducing the spectral valley energy of said input speech signal when said spectrum tilt of said input speech signal is equivalent to said spectrum tilt of said background noise; and

an object for applying an inverse filter to said input speech signal when said spectrum tilt of said input speech signal is not equivalent to said spectrum tilt of said background noise, wherein said inverse filter is an inverse of the z-domain representation of said background noise.

10. The apparatus of claim 9, wherein said input speech signal comprises a plurality of sub-frames processed in sequence.

11. The apparatus of claim 9, wherein said gain is adaptively based on characteristics of said input speech.

12. The apparatus of claim 9, wherein said background noise model is a first order model.